

Date: Thu, 9 Jun 94 04:30:30 PDT
From: Ham-Homebrew Mailing List and Newsgroup <ham-homebrew@ucsd.edu>
Errors-To: Ham-Homebrew-Errors@UCSD.Edu
Reply-To: Ham-Homebrew@UCSD.Edu
Precedence: Bulk
Subject: Ham-Homebrew Digest V94 #156
To: Ham-Homebrew

Ham-Homebrew Digest Thu, 9 Jun 94 Volume 94 : Issue 156

Today's Topics:

900mhz transciever
CORRECTION! RS 49MHz to 6m (2 msgs)
Help indentify 2SC2694
Optoisolators Used to key radio.
RADIO-TO-TELEPHONE INTERCONNECT CIRCUIT?
Ten-Tec Direct conversion kit -- getting on frequency
What's the DAC for? (3 msgs)

Send Replies or notes for publication to: <Ham-Homebrew@UCSD.Edu>

Send subscription requests to: <Ham-Homebrew-REQUEST@UCSD.Edu>

Problems you can't solve otherwise to brian@ucsd.edu.

Archives of past issues of the Ham-Homebrew Digest are available
(by FTP only) from UCSD.Edu in directory "mailarchives/ham-homebrew".

We trust that readers are intelligent enough to realize that all text
herein consists of personal comments and does not represent the official
policies or positions of any party. Your mileage may vary. So there.

Date: Tue, 7 Jun 1994 17:03:26 GMT
From: ihnp4.ucsd.edu!usc!howland.reston.ans.net!gatech!swrinde!cs.utexas.edu!
utnut!utcsri!newsflash.concordia.ca!CC.UMontreal.CA!poly-vlsi!
nick@network.ucsd.edu
Subject: 900mhz transciever
To: ham-homebrew@ucsd.edu

In article <2t12r5\$gpm@usenet.INS.CWRU.Edu> au440@cleveland.Freenet.Edu (Kieran
Donahue) writes:

>
> How would I build a synthesized 900 mhz transciever? I
>want to build a separate unit for contesting. Are there any
>good recipies that someone could recommend or acquire for me?
>--
Kieran:

The simplest way is to use a VHF or UHF radio as an IF and build yourself a transverter. Probably the cheapest way too.

Nick

--

* Nick Ciarallo *
* SR Telecom Inc. telephone: 514-335-2429 ex: 438 *
* Microwave Group facsimile: 514-334-7783 *
* 8150 Trans Canada Hwy internet : nick@vlsi.polymtl.ca *
* St. Laurent, Quebec hamradio : ve2hot@ve2fkb.pq.can.na *
* Canada H4S-1M5 *

* Accept no substitutes, *REAL* ham radio lives on 220 MHz! *

Date: 8 Jun 1994 03:00:12 GMT
From: koriel!newsworthy.West.Sun.COM!abyss.West.Sun.COM!spot!myers@ames.arpa
Subject: CORRECTION! RS 49MHz to 6m
To: ham-homebrew@ucsd.edu

In article 11120@ttinews.tti.com, sorgatz@avatar.tti.com (Erik Sorgatz) writes:
> Please note that the existing xtal in the RS 5ch 49MHz unit
>is 10.24 MHz! Substitute a 10.4 MHz xtal (its a common xtal
>used in many vdt and vtac circuits - \$1 at most surplus type
>places.) in it's place and your little ht will have 5 channels
>starting from 50.610 MHz with 15 KHz spacing.

I thought so; I calculated 10.4 MHz to move from 49.85MHz to 50.61MHz.
The channel spacing 15.23 KHz, close enough for most radios.

However, this means your second mixer injection is now 10.4 MHz, not 10.24 Mhz. This change is probably narrow enough to remain inside the passband of your first IF filter (a ceramic 10.7MHz jobby maybe as much as 300KHz wide), but means your radio is going to be most sensitive to signals 160Khz away from where you expect. In fact, I'd expect the receiver to be rather deaf at the expected receive frequency, since the 2nd IF filter is pretty narrow (< 30kHz).

In the absence of an on-frequency signal, there's probably enough signal leaking through the 2nd IF filter to demodulate the signal, but the sensitivity has to be poor. Do you retune the quad detector coil (L107), too? I'll bet that makes the audio sound better, but it won't make the radio any more sensitive.

The L7250 Sanyo synth chip uses ROM lookup tables to determine frequency, so compensating for the oddball offset of 160KHz in the IF response isn't easy. What did you do about this?

> You will have to tune the cans and the final for peak output, >but the adjustment is very slight. You might also be able to >push the freq a little with a different freq xtal, we tried a >10.7 and found it to be beyond the tuning range of the cans. So >try a 10.45 or 10.5, maybe even a 10.6 MHz would work.

Yeah, but unless I'm mistaken, you're going to run into the 2nd LO issue, too. Even worse.

> The easiest way to do this mod is to do a pair and tune them >together, this is what KB6LUZ, Dennis and I did on our initial >pair. Thereafter we used my FT726r and his Yaesu 6m rig to do >the tuneups. A longer telescopic antenna helps the range, we >picked a pair of 4 section antennas at RS. These stick out from >the top of the ht about 3-4" but are the same base diameter >as the ones supplied with the ht for 49 MHz use. I think they >are 26" as opposed to 18" long. Whatever antenna you decide to >use, tune the final to it using a field strength meter to guide >you.

I'd be interested in hearing the results of a sensitivity check on the receiver; Erik, you know where to get access to a service monitor? Let me know if I can help...

73

* Dana H. Myers KK6JQ, DoD#: j | Views expressed here are
*
* (310) 348-6043 | mine and do not necessarily *
* Dana.Myers@West.Sun.Com | reflect those of my employer
*
* This Extra supports the abolition of the 13 and 20 WPM tests *

Date: 9 Jun 94 02:21:26 GMT

From: dog.ee.lbl.gov!agate!howland.reston.ans.net!usenet.ins.cwru.edu!po.cwru.edu!
sct@ucbvax.berkeley.edu
Subject: CORRECTION! RS 49MHz to 6m
To: ham-homebrew@ucsd.edu

In article <2t3c7s\$c2v@abyss.West.Sun.COM>,
Dana Myers <myers@spot.West.Sun.COM> wrote:

> However, this means your second mixer injection is now 10.4 MHz, not
> 10.24 Mhz.

Would it be practical to add a 10.4 MHz oscillator for the PLL but retain the 10.24 MHz oscillator for the second mixer? Sorry, I don't have the schematic, so I can't see whether this is reasonable.

How about leaving the 10.24 MHz crystal in place, but use a mixer (NE602?) and filter to move the 49 MHz PLL output into the 50 MHz band? I suppose that enough filtering to avoid ~46 MHz spurious emissions would be more work and bulk than is worthwhile for the rig.

Stephen

--
Stephen Trier
sct@po.cwru.edu
KG8IF

Date: Wed, 08 Jun 1994 00:53:36 GMT
From: ihnp4.ucsd.edu!usc!howland.reston.ans.net!usenet.ins.cwru.edu!
ns.mcs.kent.edu!kira.cc.uakron.edu!malgudi.oar.net!witch!ted!
mjsilva@network.ucsd.edu
Subject: Help indentify 2SC2694
To: ham-homebrew@ucsd.edu

In article <2t0oi4\$hf4@aggedor.rmit.EDU.AU>, david@pitvax.xx.rmit.edu.au (david@pitvax.xx.rmit.edu.au) writes:
>I've had a pair of RF power transistors in the junk box for some
>time now, and thought I'd enquire as to what they are, in case it
>spurs me to do something with them!
>
>They are 2SC2694, and have the three diamonds in a circle symbol on
>them which I believe is Mitsubishi.

My Motorola RF book says these are direct equivalents to their MRF247.

These are rated at 75 watts out at 175 MHz (12 volt supply). Gain is 7.0 db min.

Let us know how your new 2m amplifier works out!

Mike, KK6GM

Date: 8 Jun 94 15:41:35 GMT
From: news-mail-gateway@ucsd.edu
Subject: Optoisolators Used to key radio.
To: ham-homebrew@ucsd.edu

Howdy,

I am interested in building an opto-isolated interface between a PC and a radio. I want to be able to key the transmitter via the LPT port, such as done by many logging programs. Specifically, I want to eliminate RF-feedback problems that occur when operating at high (1.5Kw) power levels.

Can someone suggest a circuit and/or specific device (part#) to accomplish the above task.

Thanx & 73,
Walt - K2WK

73 de Walt Kornienko - K2WK Frankford Radio Club
Internet: waltk@pica.arm.mil Snail: RR1 Box 919, Lafayette, NJ 07848
DX PacketCluster: K2WK > W3MM Packet: K2WK@NX2P.NJ.USA.NA

Date: 8 Jun 1994 00:49:43 GMT
From: ihnp4.ucsd.edu!swrinde!gatech!usenet.ins.cwru.edu!po.cwru.edu!
sct@network.ucsd.edu
Subject: RADIO-TO-TELEPHONE INTERCONNECT CIRCUIT?
To: ham-homebrew@ucsd.edu

In article <1994Jun7.135656.1@aa.wl.com>,
NONE <pennind@aa.wl.com> wrote:
> CAN ANYONE DIRECT ME ON HOW TO FIND A CIRCUIT THAT WILL INTERCONNECT MY BASE
> STATION TO THE TELEPHONE AND ALLOW ME TO MAKE AUTOPATCHES WITH MY HT...

THE ARRL HANDBOOK HAS A CHAPTER THAT DISCUSSES HOW PHONE PATCHES OPERATE.
IT IS QUITE INTERESTING, AND INCLUDES SOME INFORMATION ON A BASIC, MANUALLY
OPERATED PHONE PATCH.

An autopatch is a more complicated beast and would take a lot of work to build. You are probably better off buying a commercially made autopatch, or simpler yet, join a local repeater club that has an autopatch on their machine.

If you decide to set up your own autopatch, make sure you take care of compliance with the FCC rules. The issues (particularly control issues)

get complicated fast when dealing with autopatches.

Stephen

--
Stephen Trier
sct@po.cwru.edu
KG8IF

Date: 7 Jun 1994 18:22:42 GMT
From: newsgate.watson.ibm.com!watnews.watson.ibm.com!vinod@uunet.uu.net
Subject: Ten-Tec Direct conversion kit -- getting on frequency
To: ham-homebrew@ucsd.edu

I recently got the Ten-Tec direct conversion receiver (kit #1056), and put it together for 80m. (The kit comes with parts for all the different bands). My intention was to use it for receiving W1AW code practice. After I put the kit together, I was able to receive various signals, but mostly seemed to be digital modes, and perhaps couple of beacons, but no W1WA.

After checking that I had used all the component values they specified for the frequency determining circuits, yesterday I went over to a friends house, to check it out with an oscilloscope. After playing around with all the adjustments they specified, we found that the lowest we could go was about 3.8MHz, and the highest was about 4.3MHz. No wonder I could not get W1AW, which is at 3.5815 or so...[The specs say that the 80m module should get me between 3.5 and 4.0MHz].

Now, I am new to this stuff..so, if you can tell me what I should try to get this fixed, I would appreciate it very much. The receiver uses an NEC612AN mixer-oscillator (similar to NE602 according to the documentation).

Here is what I plan to try:

1. Resolder all the frequency determining components.
2. Try changing (ie increasing the capacitance) of the circuit which determines the local oscillator frequency. (Currently C3=150p, C4=150pf, L2=8.0uH, L3=8.2uH, C3 connected between pins 6 and 7, C4 connected from pin 7 to gnd, and L2 and L3 connected in series from pin 6 to Gnd)

through a 0.1pf capacitaor, and a varactor tuning circuit connected to the top of L2 ie aafter the 0.1pf cap).

3. Any other suggestions?

Many thanks in advance for any help, advice you can offer.

--vinod
email: vinod@watson.ibm.com

Date: 9 Jun 94 00:13:27 GMT
From: sdd.hp.com!hp-pcd!hpcvsnz!tomb@hplabs.hpl.hp.com
Subject: What's the DAC for?
To: ham-homebrew@ucsd.edu

Darren Leigh (dlleigh@frank.harvard.edu) wrote:
: I've been reading Ulrich Rohde's article in the June issue of QST. In
: figure 14 he shows a block diagram for a DDS-driven PLL. The DDS feeds
: a DAC, which gets filtered and then hard limited. This hard limited
: signal (a square wave) is then used as an input to the PLL.

: My question is: what's the DAC for? Why not just drive the PLL with the
: MSB of the DDS. It's seems like a waste to make a nice filtered analog
: signal from the DDS output and then just make a square wave again.

: My guess is that is has something to do with decreasing phase noise, but
: I don't see how it does that. Does anyone have any ideas?

I think this question has been answered plenty well enough by other posters, but to extend the thought, have a look at "Reducing Spurious Output of NCO's" on page 48 of June 1994 NASA Tech Briefs. The article talks about a (NASA-patented) technique for adding dithering to an NCO (DDS system) to reduce the level of the spurs.

Date: Tue, 7 Jun 1994 20:52:57 GMT
From: ihnp4.ucsd.edu!usc!elroy.jpl.nasa.gov!lll-winken.llnl.gov!noc.near.net!
usenet.elf.com!rpi!psinntp!arrl.org!jbloom@network.ucsd.edu
Subject: What's the DAC for?
To: ham-homebrew@ucsd.edu

Darren Leigh (dlleigh@frank.harvard.edu) wrote:
: I've been reading Ulrich Rohde's article in the June issue of QST. In
: figure 14 he shows a block diagram for a DDS-driven PLL. The DDS feeds
: a DAC, which gets filtered and then hard limited. This hard limited

: signal (a square wave) is then used as an input to the PLL.
:
: My question is: what's the DAC for? Why not just drive the PLL with the
: MSB of the DDS. It's seems like a waste to make a nice filtered analog
: signal from the DDS output and then just make a square wave again.
:
: My guess is that is has something to do with decreasing phase noise, but
: I don't see how it does that. Does anyone have any ideas?

Let me see if I can explain it. The concern here is quantization noise. Consider a sampled sine wave signal, say a 1.1 MHz signal sampled at a 4-MHz rate. The sample values would be:

```
x(k) = sin(2*pi*k*1.1/4)  
so,  
x(0) = 0  
x(1) = 0.987688341  
x(2) = -0.309016994  
x(3) = -0.891006524  
x(4) = 0.587785252
```

and so on.

Actually, there are more digits to the right of the ones shown: an infinite number of digits, in fact. But any DAC can have only a finite number of bits, so for each value shown above, the DAC can output only an approximation of the desired value. So, the DAC is usually going to be outputting a voltage slightly different from the correct one. This difference, or error, is noise. (In the case of a DDS, the noise shows up not as random "white" noise, but at discrete frequencies--spurs.) The larger the difference between the correct sample value and the value the DAC has to output, the more noise. And this is controlled by the number of bits of the DAC.

So, if we remove the DAC and just use the MSB of the DDS, we'll have huge errors for most of the output samples, which will show up as huge spurs. For example, thge samples above would have to be:

```
x(0) = 1  
x(1) = 1  
x(2) = -1  
x(3) = -1  
x(4) = 1
```

(Counting any sample with the MSB = 0 as a 1 voltage and any sample with MSB = 1 as a -1.) Not a good thing; the error (noise) is as large as the signal! And you can't count on the bandpass filter to remove the spurs generated this way, since some of them may occur close in frequency to the output signal.

Hope that helps.

--

Jon Bloom KE3Z jbloom@arrl.org

Date: 8 Jun 94 20:50:51 GMT
From: agate!howland.reston.ans.net!usc!nic-nac.CSU.net!charnel.ecst.csuchico.edu!
psgrain!news.tek.com!mdhost!bens@ucbvax.berkeley.edu
Subject: What's the DAC for?
To: ham-homebrew@ucsd.edu

In article <1994Jun4.143517.32939@husc14.harvard.edu>, dlleigh@frank.harvard.edu
(Darren Leigh) writes:

|> I've been reading Ulrich Rohde's article in the June issue of QST. In
|> figure 14 he shows a block diagram for a DDS-driven PLL. The DDS feeds
|> a DAC, which gets filtered and then hard limited. This hard limited
|> signal (a square wave) is then used as an input to the PLL.
|>
|> My question is: what's the DAC for? Why not just drive the PLL with the
|> MSB of the DDS. It's seems like a waste to make a nice filtered analog
> signal from the DDS output and then just make a square wave again.

The DDS works by keeping track of the number of Reference clock pulses required to generate a cycle of the output frequency. Quite often this is not an integer, and the Accumulator overflows at different intervals to give you an average frequency that matches what you want.

If only the MSB of the phase accumulator is used then the edge of your output signal is jumping back and forth to give you this average frequency. The DAC with the Sine table has an interpolation effect to give you less jitter (a deterministic form of phase noise), as additional information about the correct zero-crossing time is embedded in the other bits of the accumulator. As a result the edge jumping is reduced to some small percentage determined by the phase-quantization (how many points along a cycle do you have entries in the table) and DAC resolution (how many different levels between + and - peaks of the sine wave). Now you can sense the zero crossings and recreate a square wave with a much more stable edge.

Benjamin Sam
yauwah.sam@tek.com

End of Ham-Homebrew Digest V94 #156
